S04 user guide

1. Welcome

Congratulations on becoming an owner of IP-Phone S04! You made an excellent choice and we hope you will enjoy all its capabilities.

IP-Phone S04 offers a rich set of functionality and superb sound quality at ultra affordable price. It is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

2. Product brief

2.1. Key Features

- Support SIP 2.0, TCP/UDP/IP, RTP/RTCP, HTTP, HTTPS, ICMP, ARP, DHCP, NTP/SNTP, TFTP protocols.
- Support STUN and symmetric RTP for NAT.
- Interoperable with various 3rd party SIP end user device, Proxy/Registrar Server, and gateway products.
- Advanced 32bit embed MCU + DSP technology to ensure superior audio quality.
- Advanced adaptive jitter buffer control, packet delay and loss concealment technology.
- Support popular vocoders including G.723, G.729A, G.711(a-law and u-law), G.726(40K/32K/24K/16K).
- Support standard voice feature such as Caller ID Display, Call Waiting, Hold, Transfer, in-band and out-of-band DTMF, three way conference.
- Support Silence Suppression, VAD(Voice Activity Detection), CNG(Comfort Noise Generation), Line Echo Cancellation(G.168), and AGC(Automatic Gain Control)
- PSTN pass through port, with auto switch between PSTN and VOIP on inbound call and manual switch on outbound call. It becomes PSTN phone when unit power is off.

2.2. Hardware Overview

IP-Phone S04 has following interfaces for networking, visual status indication, and power:

1. LAN: one Ethernet 10baseT RJ-45 jack interface.
2. PHONE: One phone (FXS) RJ-11 jack interface.
3. LINE: One line (FXO) RJ-11 jack interface
4. One 5V power adapter interface. It is used to connect to the power adapter going with the unit. **If you connect the unit to some other power adapter, we are not guarantee the unit works properly.**

2.3. Operation Status

1. The unit is set to PSTN connection status by default. When phone is hooked, the unit is PSTN connection status. When an inbound VOIP call coming, the unit will automatically change the connection status to VOIP immediately.
2. Using PSTN and VOIP switch button (KEY"VOIP") selects outbound call connection path. The default is PSTN connection.
3. During PSTN call, an inbound VOIP call coming, VOIP connection status indicator will blink. Click the switching button (KEY"VOIP") to receive the VOIP call. The PSTN call is terminated when switching button is clicked.
4. During VOIP call, an inbound PSTN call coming, PSTN connection status indicator will blink. Click the switching button (KEY"VOIP") to receive the PSTN call. The VOIP call is terminated when switching button is clicked.

3. Installation

Please make sure you have the following items in the product package you received,

1. IP-Phone S04 IP phone
2. 5V power supply

In addition, you need,

1. A Cat-5 Ethernet cable
2. An Ethernet port connecting to IP networks

Please follow the instruction below for IP phone setup:

1. Insert one end of Cat-5 Ethernet cable to "LAN" port on the back of IP phone and the other end to Ethernet port connecting to IP networks.
2. Insert one end of RJ-11 telephone wire to the "Line" port on the back of IPP, and the other end to the telephone jack on the wall.
3. Insert 5V power supply cable into the power plug of IP phone.
4. Check and confirm all connections are correct.

Now you can power on the 5V power supply.

4. Setup Guide

It is possible that this product has been pre-setup by either a service provider or network manager. Then user's own configurable settings can be very limited. However, if end user is allowed to make changes to the system and service settings, the following are the instructions for your device setup.

End user needs to get the following information from his/her service provider or network manager,

- IP address if it is static, subnet and network gateway IP address, or user DHCP.

- SIP proxy server IP address and any registration information.
- User information such as user ID and password.

4.1 Setup with Keypad

Pressing “**00”to enter IP-Phone S04 setup menu. You can use keypad referencing following menu to do basic network setup. Please use web browser to do more advance setup.

4.1.1 Keypad Setup Menu

Option	Code	User Input	Notes
Enter configuration menu	**00	None	After “Setup menu”, enter option and end by “#”
Check DHCP	10	None	Announce DHCP Status on or off
Set DHCP	11	1 to enable 0 to disable	Press “#” to confirm the input
Check IP Address	20	None	
Set static IP Address	21	Enter value using numbers on the telephone pad. Use‘*’when enter a decimal point.	End input by ‘#’
Check Subnet Mask	30	None	
Set Subnet Mask	31	Enter value using numbers on the phone pad. Use ‘*’ for decimal point.	End input by ‘#’
Check Gateway	40	None	
Set Gateway	41	Enter value using number on the ， phone pad. Using ‘*’ for decimal point	End input by ‘#’.

Check primary DNS Server setting	50	None	
Set Primary DNS Server	51	Enter value using number on the phone pad. Using '*' for decimal point	End input by '#'.
Manual Reboot	9999	None	ATA will Automatically reboot After "hang up"
Manufacture Reboot	79898	Enter "1" to confirm Enter "*" to cancel	ATA will reboot and all the datas will return to the original form

4.2 Setup with Web Browser

Connecting IP-Phone S04's "LAN" port to a router. The procedure in 4.2.1 shows you how to setup your IP Phone and the menu in 4.2.2 for your reference.

4.2.1 Use Web Browser for system setup

1. After you have got your IP phone's IP address, you can use web browser for system setup within the same subnet.
2. Enter http://<Phone IP address>/, where <Phone IP Address> is the said IP address of your IP phone.
3. Web browser will show the webpage interface of the system setup menu. Choose "Admin Login" to enter administrator setup,
4. We use Free-World-Dialup (FWD) IP phone service as an example to setup network and service.

If your IP Phone is in a LAN and your LAN connects to Internet through a firewall (with NAT), your IP phone's IP address will be private. Your system settings should include the follows,

1.Choose outbound proxy server for NAT traversal

Under User Account Setup

- Set "User ID" to the number provided by your service provider. If you use FWD, here you can put your FWD number e.g. 180000. As for how to get free FWD number, please check details of http://www.fwd.pulver.com.
- Set "Password" to the password provided by your service provider.

Under Proxy Server Setup

- Set “Proxy” to the SIP proxy server IP address or domain name. In FWD case, it is “fwd.pulver.com”.
- Set “Use Outbound Proxy” to “yes”.
- Set “Outbound Proxy” to be the outbound proxy server IP address or domain name. In FWD case, it is http://fwdnat.pulver.com:5082.

At bottom of the pager:

- Click “Save Changes” to apply the changes.

2.Choose STUN server for NAT traversal

Under User Account Setup:

- Set “User ID” to the number provided by your service provider. If you use FWD, here you can put your FWD number e.g. 180000.
- Set “Password” to the password provided by your service provider.

Under Proxy Server Setup:

- Set “Proxy” to the SIP proxy server IP address or domain name. In FWD case, it is “fwd.pulver.com”.
- Set “Use Outbound Proxy” to “no”.

Under NAT Support:

- Set “NAT Mapping Enable” to “yes”.
- Set “NAT Keep Alive Enable” to “yes”.
- Set “Substitute VIA addr” to “yes”.
- Set “Stun Enable” to “yes”.
- Set “STUN server” to be one of the public STUN server IP address or domain name. For an example, http://stun01.sipphone.com.

At bottom of the pager:

- Click “Save Changes” button to apply all the changes.

If your IP Phone is connected to the Internet directly, your IP phone’s IP address will most likely be public IP address. Your setting should be,

Under User Account Setup:

- Set “User ID” to the number provided by your service provider. If you use FWD, here you can put your FWD number e.g. 180000.
- Set “Password” to the password provided by your service provider.

Under Proxy Server Setup:

- Set “Proxy” to the SIP proxy server IP address or domain name. In FWD case, it is “fwd.pulver.com”.

At bottom of the pager:

- Click “Save Changes” button to apply all the changes.

4.2.2 Setup Menu

Network Information
Under this tab, the current network information of unit is presented.

Product Information
Under this tab, the current product information of unit is presented.

System Status
Under this tab, the states and statistics information, which are collected by unit during normal operation, are presented. Line maintains up to 2 calls: Call 1 and 2.

Hook State	State of the hook switch
Registration State	Registration state of the line
State	State of the call
Tone	Tone playing for this call
RTP Packets Sent	Total number of RTP packets sent
RTP Bytes Sent	Total number of RTP bytes sent
RTP Packets Recv	Total number of RTP packets received
RTP Bytes Recv	Total number of RTP bytes received
SIP Messages Sent	Total number of SIP messages sent
SIP Bytes Sent	Total number of SIP bytes sent
SIP Messages Recv	Total number of SIP messages received
SIP Bytes Recv	Total number of SIP bytes received

Password Setup

Parameter Name	Function	Default
Admen Password	The password for administrator	
User Password	The password for User	

Note: the above password is only for getting in “Setup Menu”

Network Setup

Parameter Name	Function	Default
DHCP	Enable or Disable DHCP	yes
Static IP	Static IP address of the unit, which will take effect if DHCP is disabled	0.0.0.0
HostName	Host Name of ATA	
Domain	The network domain of the unit	
NetMask	The NetMask used by unit when DHCP is disabled	255.255.255.0
Gateway	The default gateway used by unit when DHCP is disable	0.0.0.0

Primary DNS	DNS server used by the unit in addition to DHCP supplied DNS servers if DHCP is enable; when DHCP is disable, this will be the primary DNS server
Secondary DNS	DNS server used by the unit in addition to DHCP supplied DNS servers if DHCP is enable; when DHCP is disable, this will be the secondary DNS server

SIP and RTP

Parameter Name	Function	Default
SIP port	SIP message listening port and transmission port	5060
Ext SIP Port	External port to substitute for the actual SIP port of the unite in all outgoing SIP Messages. If “0” is specified, no SIP port substitution is performed.	
RTP Port Min	Minimum port number for RTP transmission and reception	16580
RTP Port Max	Maximum port number for RTP transmission and reception	16680
RTP Packet Size	Packet size in sec. Value must be multiple of 0.01s(0.01 to 0.16)	0.020

NAT Support

Parameter Name	Function	Default
NAT Mapping Enable	Enable the use of externally mapped of IP address and SIP/RTP ports in SIP message.	yes
NAT Keep Alive Enable	If it is sent to “yes”, “NAT Keep Alive Msg” is sent periodically.	yes
Handle_VIA_reveived	If set to “yes”, the unit will process the “received” parameter in the VIA header inserted by the server in a response to any one of its request. Else the parameter is ignored	no
Handle_VIA_rport	If set to “yes”, the ATA will process the “rport” parameter in the VIA header inserted by the server in a response to any one of its request. Else the parameter is ignored.	no
Insert VIA reveived	Insert received parameter in VIA header in SIP responses if received from IP and VIA sent-by IP differ	no

Insert VIA rport	Insert rport parameter in VIA header in SIP responses if received-from port and VIA sent-by port differ	no
Substitute VIA addr	Use nat-mapped IP: port value in VIA header	no
Send Resp To Src Port	Send response to the request source port instead of the VIA sent-by port	no
STUN Enable	Enable the use of STUN to discover NAT mapping	no
STUN Sever	STUN server to contact for NAT mapping discovery	
Ext IP	External IP address to substitute for the actual IP address of the unit in all outgoing SIP Messages. If “0.0.0.0” is specified, no IP Address substitution is performed.	
Ext RTP Port Min	External port mapping of “RTP Port Min”. If value is non-zero, the RTP port number in all outgoing SIP messages is substituted by the corresponding port value in the external RTP port range	

Proxy Server Setup

Parameter Name	Function	Default
Proxy	SIP Proxy Server for all outbound requests	
Use Outbound Proxy	Enable the use of “Outbound Proxy”	
Outbound Proxy	SIP Outbound Proxy Sever	
Register	Enable periodic registration with the “Proxy”	yes
Make Call Without Reg	Allow making outbound calls without successful registration by the unit. If it set to “no”, dial tone will not play unless registration is successful	no
Register Expires	Expires value in second in a “REGISTER” request.	3600
Ans Call Without Reg	Allow answering inbound calls without successful registration by the unit	no

User Account Setup

Parameter Name	Function	Default
Display Name	Subscriber’s display name to appear in caller-id	
User ID	Subscriber’s user-id	
Password	Subscriber’s password	
Auth ID	Subscriber’s authentication ID	
Use Auth ID	If it is set to “yes”, the pair “Auth ID> and	no

	"Password" are used for SIP authentication. Else the pair "User ID" and "Password" are used	

Audio Setup

Parameter Name	Function	Default
Preferred Codec	Preferred codec for all calls. However, the Actual codec used in a call still depends on The outcome of the codec negotiation Protocol.	G729a
DTMF Tx Method	Method to transmit DTMF signals to the Far end	auto
Use Pref Codec Only	Only use the preferred codec for all calls.	no
Silence Supp Enable	Enable silence suppression, if it set to "yes", silent audio frames are not transmitted	no
Echo Canc Enable	If it set to "yes", the use of echo canceller is enabled.	yes
Echo Canc Adap Enable	If it set to "yes", echo canceller to adapter is enabled	yes

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4.3 Call features

Name	Code	Description
Call return	*21	call the last caller code
Call forward	*31	forward all calls to the target specified after the activation code. Example: A with number 1000, B with number 2000, pick up A, key in *312000, then hang up. When number 1000 is called, B rings instead of A. This will stay until someone picks up A, then key in *32.
Cancel call forward	*32	cancel call forward code
Call transfer	None	example: A call B, B transfer A's call to C. 1) A dial B's number. 2) B answer, then click hang up hook. After head a dial tone, key in C's number. 3) After C answers the phone, B hang up, A's call will transfer to C.
Three way conference	None	example: A call B and C for conference.

	hook.	1) A dial B's number. 2) After B answered, A(or B)click hang up 3)After A(or B)heard a dial tone, A(or B) dial C's number. 4)After C answered, A(or B)click hang up hook again, Three way conference is on.
Call Waitting	*41 *42	Press*41"to enable Press*42 to cancel
Caller ID	*51 *52	Press*51 to cancel Press*52 to cancel

5. Additional functions

5.1. Function

- 35 inbound call log
- 5 outbound call log (number and time).
- Up to 40VIP numbers
- 3 one key dialing number
- Auto redial
- One key redial
- 3 sets clock alarm
- Speaker volume digital adjust

5.2. Function setting

Setting mode

When the unit is under hang up status, push "SET" key to enter setting mode. The functions will show on LCD. Using "UP" and "Down" key to select the function you want to set. Pushing "SET" key again to enter the function. When you finish one setting, pushing "SET" to enter next one. Using "UP" and "DOWN" key to select the data. Using "CANCEL" or "EXIT" key to exit setting mode.

5.3 Function operating

Inbound Call log search

After pick up the phone, you can use "UP" and "DOWN" key to search inbound and outbound call number. Pushing "REDIAL" key. The number shows on LCD will dial out.

Outbound Call log search
Pushing “OUT DOWN” key to enter outbound call search. Using “UP” key to search recent outbound call number. LCD shows the number one by one. Pushing “REDIAL” key. The number shows on LCD will dial out.

Auto redial
When head a busy tone after you dial out, pushing “ARD” key, the unit is going into auto redialing status.

Musical hold
After phone has been picked up, push “hold” key to enter musical hold, hold musical start, then put down the handset. If

- Pick up handset again, musical stop, it will be back handset using status.
- Push “speak” button, musical stop, it will be speak using status.
- Push “Hold” button again, it will be speak using status.

Call time
As soon as phone is picked up, the call time starts to count.

Clock alarm
When alarm is on, the alarm ring will last a minute start from the alarm time. Push any key to cancel alarm.

Speak volume adjust
Push VOL key to select 2 step speak volume, LCD will indicate.

Store and fetch number
Store number: When phone is under hang up status, key in the number, push “store” key, then push store location key (such as M1, M2, or M3).
Fetch number: Push M1, M2, or M3, the number stored in the location will dial out.

6. Product Use Environment

IP-Phone 100T is recommended to work as in-door equipment.